AMENDMENTS TO THE CLAIMS

1. (Original) A system for transmitting ~~quality~~ audio signals in a communication connection from an originating device to a destination device over an IP-based network, comprising:

a port circuit for transmitting data packets, containing encoded speech signals received from an associated originating device, to said destination device via said IP-based network;

transmit buffer ~~means~~, connected to said port circuit associated with said originating device, for storing a plurality of said data packets received from said associated originating device;

network activation means for activating said IP-based network to operate using a packet transmission protocol that ~~fails to~~ does not retransmit transmitted packets that are lost or damaged ~~packets~~; and

packet retransmission means, operable ~~independent~~ independently of said packet transmission protocol and responsive to a transmitted packet being lost or damaged, for activating said port circuit to retrieve ~~a lost or damaged~~ the packet from said transmit buffer means for retransmission to said destination device.

2. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 1 wherein said packet retransmission means comprises:

packet error detection means, connected to said destination device, for generating an indication that identifies a missing packet; and

means for transmitting a signal to said port circuit associated with said originating device requesting retransmission of said identified ~~missing~~ packet.

3. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 1 further comprising:

transmit buffer control means for transmitting a signal to said port circuit associated with said originating device to regulate ~~the~~ size of said transmit ~~means~~ buffer.

4. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 1 further comprising:

jitter buffer management means for regulating a size of a jitter buffer associated with said destination device as a function of at least one of: network transmission delay, speed of

processing received packets, time required to identify ~~the~~ absence of a packet in a sequence of received packets, and time required to receive a retransmitted packet.

5. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 1 further comprising:

application detection means for determining that said communication connection serves a speech-based application that requires high quality audio signals.

6. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 5 further comprising:

network control means, responsive to said application detection means, for activating said IP-based transmission medium to transmit said high quality digital encoded speech signals without transcoding.

7. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 5 further comprising:

process disabling means, responsive to ~~the~~ conclusion of operation of said speech-based application, for disabling operation of said packet retransmission means.

8. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 5 wherein said application detection means comprises:

destination device identification means for determining ~~the~~ presence of a destination device on said communication connection that requires high quality audio signals.

9. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 5 wherein said application detection means comprises:

registration process detection means for determining ~~the~~ presence of a subscriber identification process at said destination device.

10. (Currently amended) The system for transmitting ~~quality~~ audio signals of claim 9 further comprising:

process disabling means, responsive to ~~the~~ conclusion of operation of said subscriber identification process, for disabling operation of said packet retransmission means.

11. (Currently amended) A method for transmitting ~~quality~~ audio signals in a communication connection from an originating device to a destination device over an IP-based transmission medium, comprising:

transmitting data packets, containing encoded speech signals received from said originating device, from a port circuit serving said originating device to said destination device via said IP-based ~~network~~ transmission medium;

storing, in a transmit buffer connected to said port circuit, a plurality of said data packets received from said associated originating device;

activating said IP-based network to operate using a packet transmission protocol that ~~fails to~~ does not retransmit lost or damaged packets; and

activating, independent of said packet transmission protocol and in response to a transmitted packet being lost or damaged, said port circuit to retrieve ~~a lost or damaged~~ the packet from said transmit buffer for retransmission to said destination device.

12. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 11 wherein said step of activating said port circuit comprises:

generating an indication that identifies a missing packet; and

transmitting a signal to said port circuit associated with said originating device requesting retransmission of said identified ~~missing~~ packet.

13. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 11 further comprising:

transmitting a signal to said port circuit associated with said originating device to regulate ~~the~~ size of said transmit buffer.

14. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 11 further comprising:

regulating a size of a jitter buffer associated with said destination device as a function of at least one of: network transmission delay, speed of processing received packets, time required to identify ~~the~~ absence of a packet in a sequence of received packets, and time required to receive a retransmitted packet.

15. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 11 further comprising:

determining that said communication connection serves a speech-based application that requires high quality audio signals.

16. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 15 further comprising:

activating, in response to said step of determining, said IP-based transmission medium to transmit said high quality digital encoded speech signals without transcoding.

17. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 16 further comprising:

disabling, in response to ~~the~~ conclusion of operation of said speech-based application, operation of said step of activating said port circuit to retransmit transmitted packets that are lost or damaged ~~packets~~.

18. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 16 wherein said step of determining comprises:

determining ~~the~~ presence of a destination device on said communication connection that requires high quality audio signals.

19. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 16 wherein said step of determining comprises:

determining ~~the~~ presence of a subscriber identification process at said destination device.

20. (Currently amended) The method for transmitting ~~quality~~ audio signals of claim 19 further comprising:

disabling, in response to ~~the~~ conclusion of operation of said subscriber identification process, operation of said step of activating said port circuit to retransmit transmitted packets that are lost or damaged ~~packets~~.

21. (New) A method of operating a buffer mechanism that is activated when there is an unavoidable pause in delivery of data to a receiving device comprising:

detecting when a pause in data delivery is necessary;

determining an effect that pauses at specific points in the delivery of data would have on performance of the receiving device; and

managing buffering and subsequent delivery of data to the receiving device, such that pauses in the data delivery occur at locations that would have a minimal impact on the performance of the receiving device.

22. (New) The method of 21 wherein:

the data consist of speech inputs to an automatic speech recognition resource; and

said step of managing the buffering implements pauses in the delivery of speech to the resource between words rather than within words.

23. (New) The method of 21 wherein:

the data consist of speech inputs to an automatic speech recognition resource; and

said step of managing the buffering implements pauses in the delivery of speech to the resource between phrases rather than within phrases.

24. (New) The method of 21 wherein:

the data consist of speech inputs to an automatic speech recognition resource; and

said step of managing the buffering implements pauses in the delivery of speech to the resource between commands rather than within commands.

25. (New) The method of 21 wherein:

the data consist of speech inputs to a voice-recording resource; and

said step of managing the buffering implements pauses in the delivery of speech to the resource between words rather than within words.

26. (New) The method of 21 wherein:

the data consist of speech inputs to a voice-recording resource; and

said step of managing the buffering implements pauses in the delivery of speech to the resource between phrases rather than within phrases.

27. (New) The method of 21 wherein:

the data consist of tonal inputs to a tone-detection resource; and

said step of managing the buffering implements pauses in the delivery of audio tones to the resource between tones, rather than within tones.

28. (New) The method of 21 wherein:

the data consist of audio signals in which the duration of individual signals is important; and

said step of managing the buffering implements pauses in the delivery of signals to the resource which do not occur within time-sensitive signal components.

29. (New) The method of 21, wherein:

the data consist of TTY/TDD characters; and

said step of managing the buffering implements pauses in the delivery of characters to the resource between individual characters rather than within characters.

30. (New) A method for transmitting data signals in a communication connection from an originating device to a destination device over an IP-based network, comprising:

transmitting data packets from a first communication device to a second communication device via said IP-based network using a first transmission protocol that does not retransmit transmitted packets that are at least one of lost and damaged;

determining that network performance of said IP-based network is insufficient to transmit quality data signals using the first transmission protocol; and

changing from transmitting data packets using a first transmission protocol to transmitting data packets using a second transmission protocol that provides for retransmission of transmitted packets that are at least one of lost and damaged.

31. (New) The method for transmitting data signals of claim 30 wherein said step of switching comprises:

generating an indication that identifies a missing packet; and

transmitting a signal to said first communication device requesting retransmission of said identified packet.

32. (New) The method for transmitting data signals of claim 31 further comprising:

transmitting a signal to said first communication device to regulate size of said transmit buffer.

33. (New) The method for transmitting data signals of claim 31 further comprising:

regulating a size of a jitter buffer associated with said second communication device as a function of at least one of: network transmission delay, speed of processing received packets, time required to identify absence of a packet in a sequence of received packets, and time required to receive a retransmitted packet.

34. (New) The method for transmitting data signals of claim 31 further comprising:

determining that said communication connection serves a speech-based application that requires high quality audio signals.

35. (New) The method for transmitting data signals of claim 34 further comprising:

activating, in response to said step of determining, said IP-based network to transmit said high quality digital encoded speech signals without transcoding.

36. (New) The method for transmitting data signals of claim 35 further comprising:

disabling, in response to the conclusion of operation of said speech-based application, operation of said step of activating said first communication device to retransmit transmitted packets that are lost or damaged.

37. (New) The method for transmitting data signals of claim 35 wherein said step of determining comprises:

determining presence of a second communication device on said communication connection that requires high quality audio signals.

38. (New) The method for transmitting data signals of claim 35 wherein said step of determining comprises:

determining presence of a subscriber identification process at said second communication device.

39. (New) The method for transmitting data signals of claim 38 further comprising:

disabling, in response to conclusion of operation of said subscriber identification process, operation of said step of activating said port circuit to retransmit transmitted packets that are lost or damaged.

40. (New) A method for transmitting data packets in a communication connection from a transmit buffer to a destination system via an IP-based communication network, comprising:

providing a transmit buffer for temporary storage of data packets to be sent across an IP-based communication network;

storing in said transmit buffer at least one data packet that is transmitted across said IP-based communication network; and

varying a size of said transmit buffer based on input from at least one of said IP-based communication network and a destination system which is on said communication connection and connected to said IP-based communication network.

41. (New) The method for transmitting data packets of claim 40 wherein:

said input comprises at least one of: transmission delay in said IP-based communication network, speed of processing received data packets, time required to identify absence of a data packet in a sequence of received data packets, and time required to receive a transmitted data packet.

42. (New) The method for transmitting data packets of claim 40 further comprising:

transmitting data packets from said transmit buffer to said destination system via said IP-based network using a first transmission protocol that does not retransmit transmitted packets that are at least one of lost and damaged;

determining that network performance of said IP-based network is insufficient to transmit quality data signals using said first transmission protocol; and

changing from transmitting data packets using said first transmission protocol to transmitting data packets using a second transmission protocol that provides for retransmission of transmitted packets that are at least one of lost and damaged.

43. (New) The method for transmitting data signals of claim 42 wherein said step of determining comprises:

generating, at said destination system, an indication that identifies a missing packet; and

transmitting a signal to said transmit buffer requesting retransmission of said identified packet.

44. (New) The method for transmitting data signals of claim 42 wherein said step of determining comprises:

determining presence of a destination system on said communication connection that requires high quality audio signals.